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(54) Transversal filters

(57) An adaptive transversal filter for use in rejecting interference comprises: a tapped delay line having a central tap point (1) and further tap points (2A, 2R; 3A, 3R; 4A, 4R; 5A, 5R) in pairs symmetrically disposed with respect to the central tap point; means (10—13, 14) for weighting and then summing the signals from the further tap points, the two signals from each pair of points being equally weighted; means (16) for subtracting the weighted sum from the signal received from the central tap point; and control means (17—20) for estimating the correlation between the output signal from the subtraction means and the input signal and deriving from these correlations weighting coefficients for the weighted sum.

Because the weighting coefficients are applied in a symmetrical way relative to the central tap point of the delay line, no phase distortion is introduced by the filter. The filter may be used to remove noise from a comparatively well correlated and stable signal, or to remove highly correlated interference from a relatively uncorrelated signal, eg a speech signal.

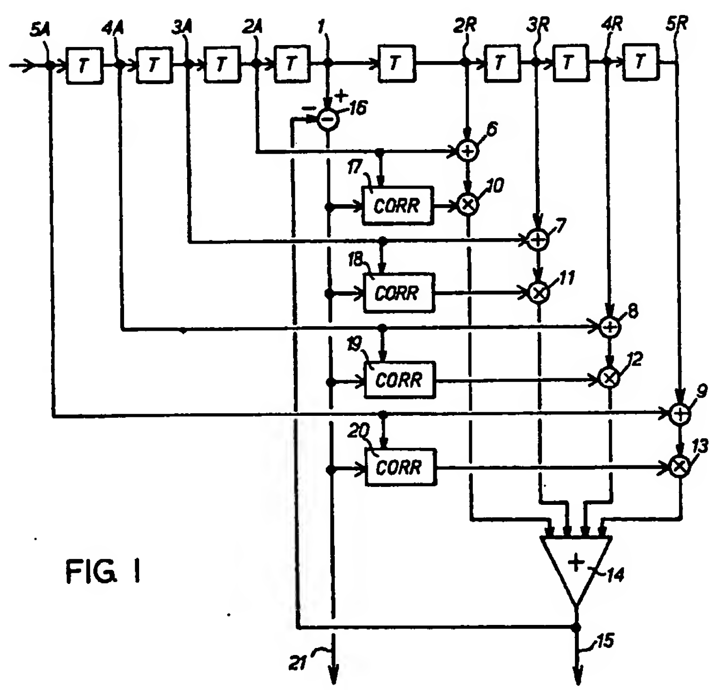

FIG. 1

The drawings originally filed were informal and the print here reproduced is taken from a later filed formal copy.

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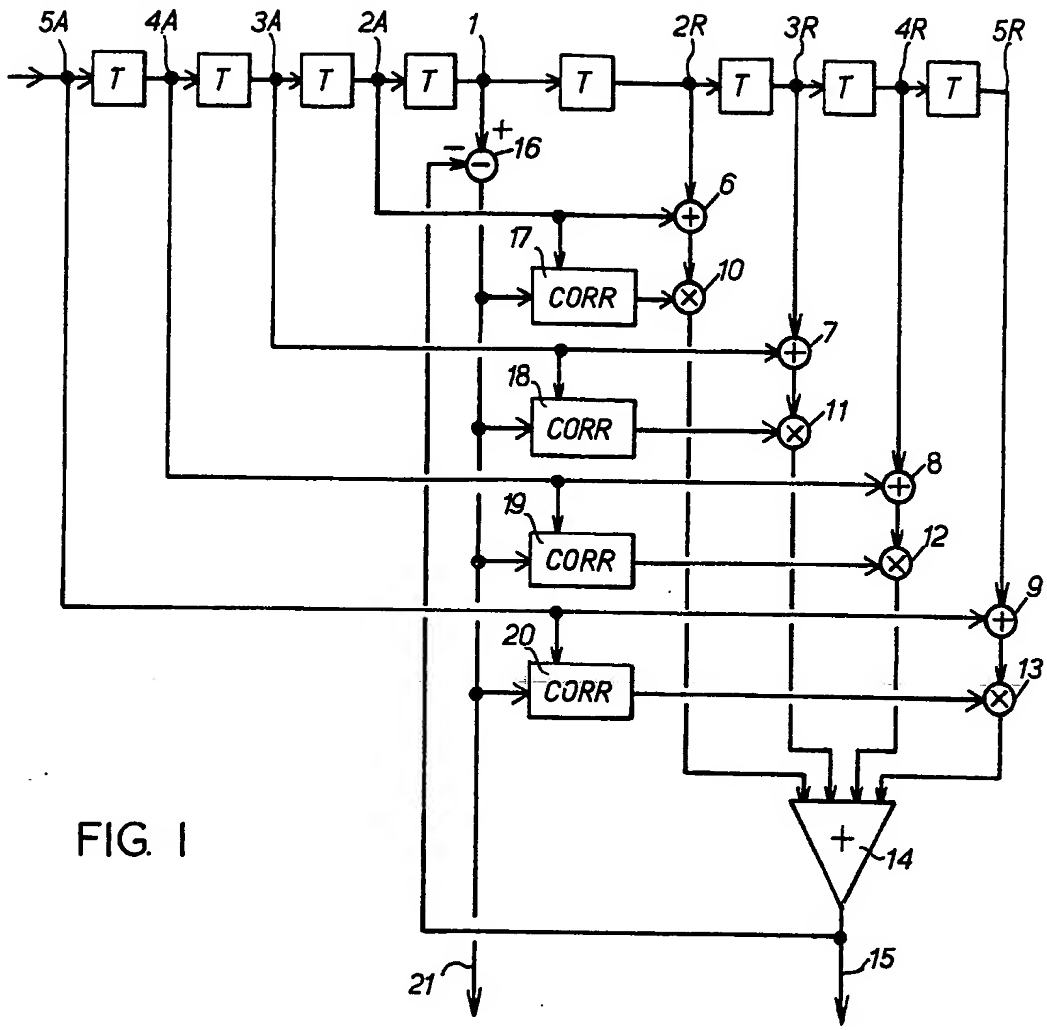
5A
4A
3A
2A
1
2R
3R
4R
5R
T
+
−
16
6
17
CORR
10
7
18
11
8
19
12
9
20
13
14
15
21

FIG. 1

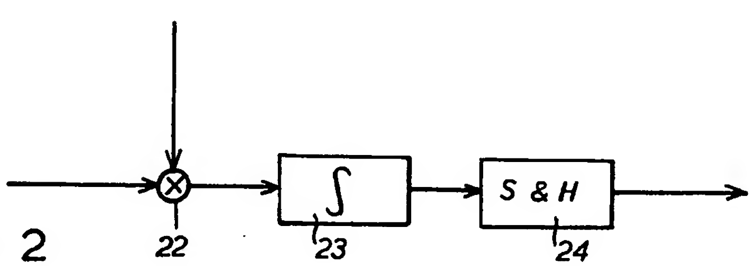
S & H
22
23
24

FIG. 2

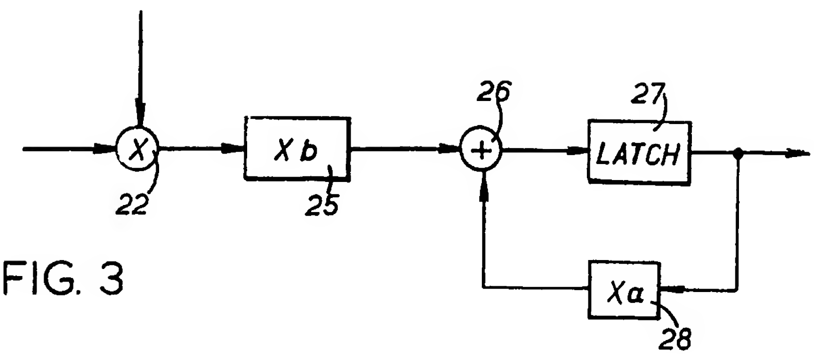

FIG. 3

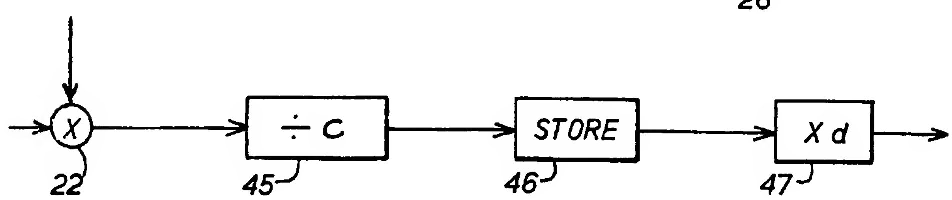

FIG. 4

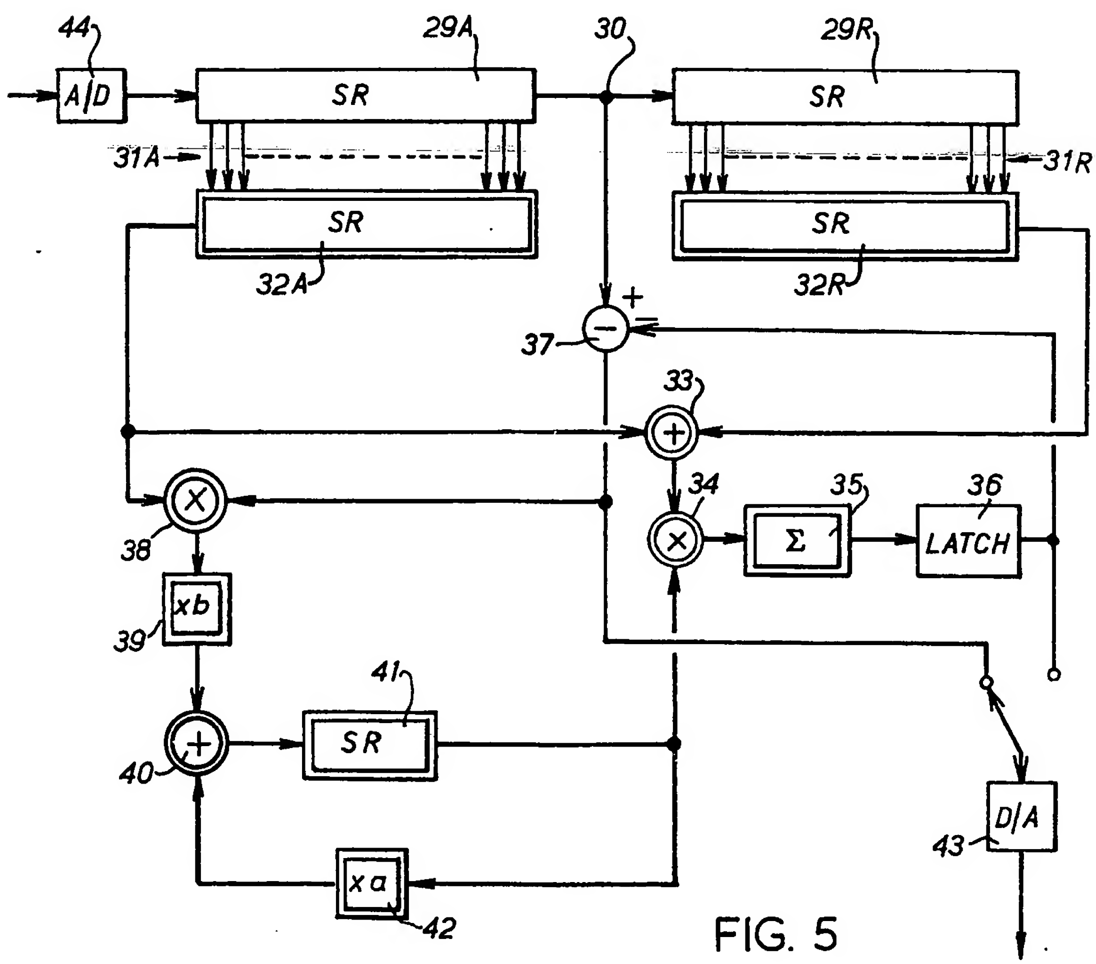

FIG. 5

SPECIFICATION

Improvements in or relating to Adaptive Transversal Filters

This invention relates to adaptive transversal filters.

Transversal filters are known in connection with a variety of applications. The principle of a transversal filter is that a signal, either an analogue signal or a digitised version of an analogue signal, is passed along a tapped delay line and the values of the signal at the taps are added together with respective weighting coefficients to give an output signal. The delay line is conveniently a clocked delay line, such as a shift register. In an adaptive transversal filter the weighting coefficients are derived or at least adjusted automatically by the filter itself.

A known type of adaptive transversal filter is the minimum-phase filter. In the minimum-phase filter a series of output signal samples y_i is derived from a series of input signal samples x_i according to

$$y_i = x_i - z_i$$

where

$$z_i = \sum_{n=1}^{N} k_n x_{i-n}$$

The N quantities k_n are the weighting coefficients and they are derived by a control loop which adjusts the k_n so that the correlations of the output series y_i with the input series x_i,

$$C_n = \text{Average } (y_i\, x_{i-n})\ (1 \leqslant n \leqslant N)$$

are reduced approximately to zero. This can be done by estimating the C_n each sample period and setting each k_n equal to the respective estimated C_n multiplied by some loop gain constant A. As an example the k_n may be adjusted according to

$$k_n\,(\text{new}) = a\, k_n\,(\text{old}) + b\, y_i\, x_{i-n}$$

The gain A is then given by

$$A = \frac{b}{1-a}$$

If the loop is stable with the loop gain A infinite then at equilibrium the C_n all vanish. In practice A is generally finite but large and the C_n at equilibrium are small.

The minimum-phase filter has the property that the output signal samples y_i, when the k_n have reached equilibrium, are uncorrelated, so the output signal has a flat spectrum.

One application of the minimum-phase filter is to remove highly correlated interference from a relatively uncorrelated signal. For example speech or facsimile transmission uses a comparatively wide bandwith and the signal is relatively uncorrelated; the spectrum is not necessarily completely flat, but it fluctuates so that it does not have a stable structure. Some kinds of interference however, such as interference from frequency shift key telegraphy, are relatively highly correlated and the structure of the spectrum remains relatively stable. A minimum-phase filter will reject a highly correlated signal once the k_n have adapted to it. If the time constants of the control loop are suitably chosen the filter will adapt itself to interference of relatively stable spectrum and reject it, while passing the relatively unstable wanted signal.

It may also be possible by taking the z_i as the output signal, to remove poorly correlated or unstable noise from a relatively stable and correlated signal.

A disadvantage of the minimum-phase filter in an interference rejecting application is that it introduces phase distortion into the signal. It is an object of the present invention to provide an adaptive filter which introduces no phase distortion.

According to the present invention an adaptive transversal filter comprises a tapped delay line for receiving an input signal and deriving delayed versions thereof, and having a central tap point and further tap points in pairs symmetrically disposed with respect to the central tap point; means connected to receive signals from the tap points for forming a derived signal, the derived signal being the difference between the signal received from the central tap point and a weighted sum of the signals received from the further tap points, the two signals from each of the pairs of tap points both having equal weights in the sum; and control means for estimating correlations between the derived signal and the input signal and deriving therefrom weighting coefficients for the weighted sum.

The control means may be arranged to set each of the weighting coefficients equal to a respective estimated correlation multiplied by a loop gain constant. The control means may comprise means for forming, for each of the pairs of tap points, a time average of the instantaneous product of the derived signal and one of, or an average of both of the signals received from the tap points in the pair and setting the weighting coefficient for the signals from the pair equal to the time average multiplied by a loop gain constant. The control means may comprise a separate means for each of the pairs of tap points for forming the respective instantaneous products and time averages or alternatively it may comprise means for taking each pair in turn in a multiplex arrangement. The means for forming the derived signal may comprise a separate multiplier for each of the pairs for multiplying the signals from the tap points with the respective weighting coefficients and an adder connected to receive the outputs of the multipliers in parallel, or it may comprise means for multiplying the signals from each pair in turn by the respective weights in a multiplex arrangement.

With the present filter, since the weighting

coefficients are applied in a symmetrical way relative to the central tap point, no phase distortion is introduced by the filter. This is not the only difference in performance between the present filter and the minimum-phase filter. Assuming that the present filter could be made stable with the weights being adjusted so that the correlations between the input signal and the derived signal vanish, for example with the loop gain infinitely large, then it can be shown that if the input signal has a stable frequency spectrum the derived signal has a frequency spectrum which is the inverse of that of the input signal, having maxima where the spectrum of the input signal has a minima and vice versa. This is in contrast to the minimum-phase filter in which the output signal has a flat spectrum. The inversion of the frequency spectrum may appear at first sight to be a disadvantage, since it would appear that any low points in the frequency spectrum of the input signal are converted into high points in the frequency spectrum of the derived signal, so that while the original interference peaks are removed new ones are created. In practice however this does not happen. If the loop gain is finite the correlations between the input signal and the derived signal do not vanish and the spectrum is not exactly inverted; low points in the spectrum of the input signal remain low points in the spectrum of the derived signal, but at regions of high intensity the spectrum is approximately inverted. This means that strong narrow-band components of interference are strongly suppressed, whereas low background components are passed with little distortion. Thus the fact that to achieve stability the loop gain may need to be finite is not always a disadvantage in the present filter. Indeed in some applications a finite loop gain is to be preferred and the best value for the loop gain may be substantially lower than the maximum value for which stability can be achieved.

If instead of taking the derived signal one takes as the output signal the difference between the derived signal and the signal from the central tap point or the weighted sum of the signals from the pairs of tap points without any contribution from the central tap point (which amounts to the same thing), then the output signal contains predominantly the highly correlated stable components of the input signal and uncorrelated or unstable components are rejected. The filter can thus be used to remove noise from a comparatively well correlated and stable signal.

Some embodiments of the invention will now be described by way of example with reference to the accompanying drawings of which:

Figure 1 is a schematic circuit diagram of a filter according to the invention,

Figures 2, 3 and 4 are schematic circuit diagrams of correlators for use in the filter of Figure 1, and

Figure 5 is a schematic circuit diagram of an alternative form of filter according to the invention.

In Figure 1 a tapped clocked delay line is represented by eight identical discreet delay circuits T connected in series. There is a central tap point 1 and four pairs of tap points 2A, 2R, 3A, 3R, 4A, 4R, 5A and 5R symmetrically disposed with respect to the central tap point 1, the tap points 2A and 2R each being separated from the central tap point 1 by one delay circuit T, the tap points 3A and 3R by two delay circuits T and so on, the tap points 2A, 3A, 4A and 5A being upstream of the central tap point 1 (i.e. advanced with respect to it) and 2R, 3R, 4R and 5R downstream (retarded). The tap points 2A and 2R are connected to respective inputs of an adder 6 and the other pairs (3A, 3R), (4A, 4R), and (5A, 5R) are connected to respective adders 7, 8 and 9. The outputs of the adders 6 to 9 are each connected to first inputs of respective multipliers 10 to 13; the outputs of which are connected to an adder 14. The adder 14 thus forms in use a weighted sum of the signals from the tap points 2A, 2R to 5A and 5R, the weights being applied by the multipliers 10 to 13. Since the signals from each pair of tap points for example 2A and 2R, are added together by their respective adder (6 in the example) before the weight is applied by the respective multiplier (10 in the example), the signals from each pair of tap points both have equal weights. The output 15 of the adder 14 forms one possible output of the filter, for use if it is desired to pass stable and correlated signals and to remove noise.

A subtractor 16 has a minuend input connected to the central tap point 1 and a subtrahend input connected to the output 15 of the adder 14. The output of the subtractor 16 is thus a derived signal consisting of the difference between the signal at the central tapping point 1 and the weighted sum produced by the adder 14. The output of the subtractor 16 is connected to one input of each of correlators 17, 18, 19 and 20, each of which has another input connected to a respective one of the upstream tap points 2A to 5A and an output connected to an input of the corresponding multiplier 10 to 13.

Three correlators are illustrated in Figures 2, 3 and 4. In the correlator of Figure 2, which particularly lends itself to analogue implementation, the two inputs are connected to respective inputs of a multiplier 22, the output of which is connected via an integrator 23 to the input of a sample and hold circuit 24. If the integrator 23 were perfect, in the sense that it was not leaky, the correlator would give an infinite loop gain, but in practice a leaky integrator is used, so the loop gain is finite. The integrator 23 also has some series gain or attenuation which affects the loop gain.

In the correlator of Figure 3, which particularly lends itself to digital implementation the integrator 23 and sample and hold circuit 24 of Figure 2 are replaced by a constant multiplier circuit 25 in series with an input of an adder 26, the output of which is connected to the input of a latch circuit 27 and the other input of which is connected via a constant multiplier circuit 28 to

the output of the latch circuit 27. The constant multiplier circuit 25 provides a constant series gain or attenuation b for the correlator and the constant multiplier circuit 28 provides a constant attenuation a in the feedback from the output of the latch 27 to its input. this correlator thus provides an exponentially weighted average of previous values of the product of the inputs, with a loop gain

$$A = \frac{b}{1-a}$$

The correlator of Fig. 4 again is suitable for use in digital implementation of the filter. In addition to the multiplier 22, it includes a constant division circuit 45 arranged to divide the output of the multiplier 22 by a large number c, a store 46 the content of which is continuously updated (i.e. every sample time) by the addition thereto of the output of the division circuit 45, and a constant multiplier circuit 47 arranged to periodically (e.g. every 100th sample) multiply the output of the store by a value d slightly less than unity to produce, in effect, a leaky integrator. The attack time of the filter is thus determined by the value of the number c and the decay time by the value of the constant d and the frequency with which it is multiplied with the output of the store 46.

Although in each of the above arrangements the correlation coefficients are estimated from the derived signal appearing at the output of the subtractor 16 and advanced signal samples taken from the tap points 2A to 5A, the retarded signal samples from tap points 2R to 5R, or an average of both may similarly be used. In the latter arrangement, one of the correlator input signals may (with suitable modification) be taken from a respective one of the adders 6, 7, 8, 9 instead of directly from the tap points.

The filter of Figure 1 may be implemented in analogue form, using the form of correlator shown in Figure 2, and using a bucket brigade or charge coupled delay line, or in digital form, using a multi-bit shift register as the delay line and the form of correlator shown in Figure 3 or Figure 4. In the case of digital implementations the connections shown as single lines in the Figures will be multi-bit connections. Although for the sake of clarity a filter with only four pairs of tap points has been described a larger number would be more typical in practice, as is the case with minimum-phase filters.

In the filter illustrated in Figure 1 there is a separate set of arithmetic components (adder, multiplier and correlator) for each pair of tap points. This could mean in practice that there would be about a hundred sets of arithmetic components. Figure 5 shows a digital implementation of the filter in which the arithmetical operations of correlation and forming the weighted sum are performed by a single set of arithmetic components in a multiplex arrangement. In Figure 5 connections are shown as single lines although it is to be understood that except for the input and output connections to the filter, which are for analogue signals, they are parallel multi-bit connections.

The input connection of the filter is connected via an analogue-to-digital converter 44 to a serial input of a 256-stage eight-bit-wide shift register formed from two 128-stage shift registers 29A and 29R in series. A serial output of the upstream shift register 29A forms a central tap point and taps 31A and 31R on the shift registers 29A and 29R respectively constitute pairs of tap points symmetrically disposed about the central tap point 30. In this filter there are thus 128 pairs of tap points. The shift registers 29A and 29R are clocked at the sampling rate of the filter. The taps 31A and 31R are connected to parallel input connections of a second pair of shift registers 32A and 32R respectively. The directions of shift in the shift registers 32A and 32R are opposite in the sense that one of them shifts upstream relative to the input signal and the other downstream so that the two signals from each symmetrically disposed pair of taps 31A and 31R are both delivered simultaneously to the serial outputs of shift registers 32A and 32R. The shift registers 32A and 32R are both loaded once every sample period and are then clocked at a fast rate so that the signals from all pairs of tap points 31A, 31R are output during each sample period. The shift registers 32A and 32R thus act as multiplexers. Components which operate at the fast rate are shown with double outlines in Figure 4.

The outputs of the shift registers 32A and 32R are connected to respective inputs of an adder 33 the output of which is connected to a first input of a multiplier 34. The output of the multiplier 34 is connected to the input of a cumulative adder 35 which is cleared once each sample period, the contents then being loaded into a latch circuit 36. The signal held in the latch circuit 36 is thus a weighted sum of the signals from the pairs of tap points 31A, 31R, the weights being applied by the multiplier 34.

The contents of the latch 36 are applied to a subtrahend input of a subtractor 37, the minuend input of which is connected to the central tap point 30. The output of the subtractor 37 is thus the difference between the signal at the central tap point 30 and the weighed sum held in the latch 36. The output of the subtractor 37 and the output of the shift registers 32A are connected to respective inputs of a correlator circuit comprising a multiplier 38, constant multipliers 39, 42, an adder 40 and a 128-stage multi-bit shift register 41. The correlator is similar to that shown in Figure 3 but with the latch 27 of Figure 3 replaced by the 128-stage shift register 41 so that the correlator performs 128 separate correlations in multiplex. The output of the correlator is connected to a second input of the multiplier 34.

A digital-to-analogue converter 43 is connected to the output of the latch 36 if it is required to remove uncorrelated noise and pass a stable and correlated signal, or to the output of the subtractor 37 if it is required to remove stable and

correlated interference from a relatively uncorrelated or unstable signal. In either case the output of the digital-to-analogue converter 43 forms the output of the filter.

CLAIMS

1. An adaptive transversal filter comprising a tapped delay line for receiving an input signal and deriving delayed versions thereof, and having a central tap point and further tap points in pairs symmetrically disposed with respect to the central tap point; means connected to receive signals from the tap points for forming a derived signal, the derived signal being the difference between the signal received from the central tap point and a weighted sum of the signals received from the further tap points, the two signals from each of the pairs of tap points both having equal weights in the sum; and control means for estimating correlations between the derived signal and the input signal and deriving therefrom weighting coefficients for the weighted sum.

2. A filter as claimed in Claim 1, wherein the control means is arranged to set each of the weighting coefficients equal to a respective estimated correlation multiplied by a loop gain constant.

3. A filter as claimed in Claim 2, wherein the control means is arranged to estimate said correlations by forming, for each of the pairs of tap points, a time average of the instantaneous product of the derived signal and the signal received from one of the tap points of the pair.

4. A filter as claimed in Claim 2, wherein the control means is arranged to estimate said correlations by forming, for each of the pairs of tap points, a time average of the instantaneous product of the derived signal and an average of both of the signals received from the tap points of the pair.

5. A filter as claimed in Claim 3 or Claim 4, wherein the control means is arranged to form the respective instantaneous products and time averages for each pair of tap points in turn in a multiplex fashion.

6. A filter as claimed in any preceding Claim, wherein the means for forming the derived signal is arranged to add together the two signals received from each respective pair of tap points before applying the weighting coefficient thereto, and to then sum together the summed and weighted signals from the respective pairs of tap points.

7. A filter as claimed in any preceding Claim, wherein the means for forming the derived signal is arranged to multiply the signals from each pair of tapping points in turn with the respective weighting coefficients in a multiplex fashion.

8. An adaptive transversal filter substantially as shown in and hereinbefore described with reference to Figure 1, with reference to Figure 1 in conjunction with Figure 2, Figure 3 or Figure 4, or with reference to Figure 5.

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